Amendments to the Specification:

Please replace page 10, paragraph [0026] with the following amended paragraph:

[0026] In contrast to these approaches of the past, in the commonly-owned U.S Patent No. 6,674,717, entitled "Method for Reducing Packet Loss and Increasing Internet Flow by Feedback Control", filed March 30, 2000, which is hereby incorporated by reference in its entirety, the applicants introduced a new approach to congestion control that did exploit the chaotic nature of network traffic. Under the scheme described in U.S. Patent No. 6,674,717, the end-to-end packet losses of one or more traffic streams transmitted across a congested network may be reduced by modulating the bandwidths (i.e., the inter-packet delay) of the corresponding traffic streams applied to the node(s) of the network from one or more control points along the network topology. This reduction in packet loss results in a reduction in fluctuations or variability of the controlled traffic streams, an increase in bandwidth utilization of a congested link at downstream points and a reduction in times to transmit files (e.g., to an end-user). The control points can be either upstream or downstream of one or more congestion points along the network.

Please replace page 16, paragraph [0039] with the following amended paragraph:

[0039] To discriminate traffic flows (or connections, etc.) that are congested from those that are uncongested (or relatively uncongested), the control node monitors packet loss for the traffic stream (or connection, etc.) of interest. If after a designated time period (say between 0 and 100 seconds, preferably between 30 and 100 seconds, more preferably between 50 and 100 seconds, and even more preferably between 60 and 100 seconds) no packet losses have been noted for that traffic flow (connection, etc.), the control node can declare the subject traffic flow (connection, etc.) to be uncongested. This type of monitoring (and the subsequent control) can be provided on a stream-by-stream, connection-by-connection, link-by-link, destination-by-destination or other basis. Alternatively, predictions of congestion conditions may be based on results obtained by monitoring packet round trip times in the network. A complete description of such a process is provided in co-pending and commonly owned U.S. Patent Application No. 09/854,321, entitled "Method for Determining Network Congestion and Link Capacities", filed May 11, 2001, Attorney Docket No. 003997.P010, which is incorporated herein by reference. Briefly, it has been found that there is an intimate relationship between packet loss and packet round trip time

in a network such that measurements of this round trip time can be used to determine congestion conditions. Further methods of predicting such conditions are disclosed in commonly-owned U.S. Patent Application No. 09/846,450, entitled "Method for Dynamical Identification of Network Congestion Characteristics", filed April 30, 2001, Attorney Docket No. 003997.P008, which is also incorporated herein by reference.

Please replace page 18, paragraph [0045] with the following amended paragraph:

[0045]Applying these techniques to streams transferred across uncongested network routes depends upon the ability to accurately identify congested streams. An empirical analysis of real traffic data from major Web portal sites indicates that there are two major factors in determining congested streams. The first is the inherent congestion on a particular network route that is driven by external factors such as cross traffic. An analysis of these network routes demonstrates that this congestion varies on fairly long timescales, such as tens of minutes to several hours. The second is the traffic generated by the stream itself that is traversing this congested network route. This traffic can vary on much faster and broader timescales such as tens of seconds to several hours and varies greatly from stream to stream.

Please replace page 19, paragraph [0046] with the following amended paragraph:

[0046]An extensive analysis of real-time traffic data shows that congestion as measured by the monitored streams can be discriminated from periods of no congestion by looking at time intervals in which there is no packet loss. **Figures 4A and 4B** are plots of actual Web traffic, showing variations in throughput (**Figure 4A**) and packet loss (**Figure 4B**) as functions of time. As shown in **Figure 4C**, as time intervals approaching 60 – 100 seconds are measured without significant packet loss, we can be reasonably certain that the subject traffic flow is uncongested. For large enough time intervals then, the probability that the absence of packet loss is due to statistics of the monitored stream is negligible. Those are periods where there is no congested for the associated network route. However, intervals where there is no packet loss due purely to the statistics of the monitored stream can occur in the tens of seconds. These are periods where the inherent congestion in the route may exist, but for the underlying packet loss rate, not enough

packets from the monitored stream have been transferred along this route to make it statistically likely that one or more of those packets have been dropped.